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(54) Title: METHOD OF DYNAMICALLY COMPENSATING FOR VARIABLE TRANSMISSION DELAYS IN PACKET NETWORKS
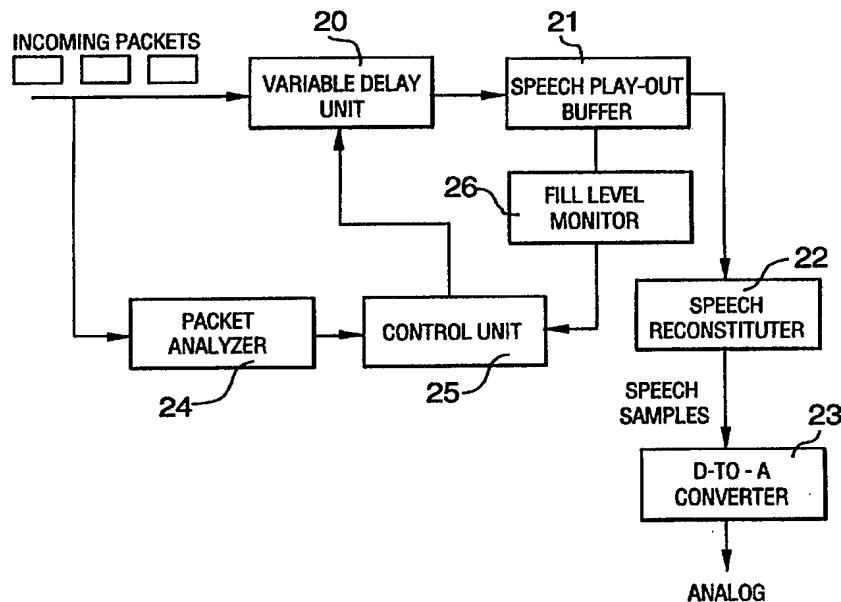
(57) Abstract

A method is described for playing out packets, such as voice or video packets, received through a packet network subject to variable transmission delays. The incoming packets are received in a delay buffer and a predetermined delay applied to the first packet of a sequence of packets. A variable delay is applied to subsequent packets to produce an appropriate constant play-out rate to reproduce the desired output. The fill level of the delay buffer is monitored and the predetermined delay applied to the first packet of a following sequence of packets adjusted to maintain the fill level within desired limits to minimize the risk of said buffer underflowing or overflowing.

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METHOD OF DYNAMICALLY COMPENSATING FOR VARIABLE TRANSMISSION DELAYS IN PACKET NETWORKS

This invention relates to a method and apparatus for dynamically compensating for variable transmission delays in packet networks, particularly voice networks, but the invention is also applicable to other networks, such as video networks. The invention is applicable in all integrated packet networks where voice or video may be carried including, for example, frame relay networks, ATM networks, PCME (packet circuit multiplication equipment), and LANs.

In this specification, reference is made throughout to "voice" packets, since this is the term normally used in the art to describe such networks, although it will be realized by one skilled in the art that such networks extend to any network capable of transmitting any form of audio whether it actually be voice or other form of reproducible sound.

Two methods have been proposed to compensate for variable transmission delays in packet voice networks.

In the first method, known as timestamping, which is used in ITU standard G.764, the accumulated variable transmission delay experienced by a voice packet is recorded in a timestamp field in the packet. Each intermediate node recognizes voice packets and adds to the timestamp field the amount of time that it took for the packet to transit the node. The receiver uses the value in the timestamp field to determine when to play-out the voice packet. Voice packets that experience little delay in the network will be delayed in the receiver longer before being played-out, and voice packets that experienced long network delays will be delayed less in the receiver. The effect is that the sum of the network delay and the delay in the receiver will

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be nearly constant for all voice packets and the voice
will be played-out at a uniform rate.

The disadvantage of this method is that intermediate
nodes must recognize voice packets and carry out special
5   processing. This makes this method incompatible with
existing networks that do not support this function.
Another disadvantage is that prior knowledge of the
maximum expected delay variation is necessary.

In the second method, known as the blind delay
10   method, a fixed delay is always added at the receiver to
the first packet of a talk sequence. The delay
corresponds to the maximum variable delay expected from
the network. This way, if the first packet experiences
minimum delay, the system compensates by adding enough
15   delay to make sure other packets, which experience more
delay, arrive before their scheduled play-out time.

The disadvantage of this method is that it may
increase the delay in the voice path beyond the optimal
value. This is because if first packet has already
20   experienced the worst case delay, the delay added will
include the worst case twice. Large delays degrade the
system performance (or may cause the system not to meet
the international standards for network delays specified
in ITU-T Recommendation G.114). Delay is of special
25   concern when speech and facsimile demodulated traffic are
mixed on the same transmission facility. Another
disadvantage is that prior knowledge of the maximum
expected delay variation is necessary.

According to the present invention there is provided
30   a method of playing out packets received through a packet
network subject to variable transmission delays,
comprising the steps of receiving incoming packets in a
buffer; applying a delay to the first packet of a
sequence of packets; applying a variable delay to
35   subsequent packets of the sequence to produce an

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appropriate constant play-out rate to reproduce the desired output; monitoring utilization of said buffer; and adjusting the delay applied to the first packet of a following sequence of packets to maintain said
5   utilization within desired limits to minimize the risk of said buffer underflowing or overflowing.

The utilization monitored can be, for example, the buffer fill level or the dwell time of packets in the buffer. Alternatively, the arrival rate of the packets
10   could be monitored.

The packets may, for example, be voice packets or video packets.

The invention (Adaptive Delay Equalization) thus uses an adaptive method to determine when to play-out
15   received packets. This method minimizes the delay applied to received packets. In a preferred embodiment, the receiver starts by applying a pre-determined delay to the first packet of a talk-spurt. The receiver delays subsequent packets by an amount appropriate to produce a
20   constant gap-free play-out rate. If the minimum number of packets in the buffer is large (i.e., the buffer is never close of under-flowing), the system slowly reduces the predetermined delay, known as the build-out delay. If packets arrive late, the build-out delay is increased in
25   order to minimize packet loss. The build-out delay adjustment can be done during speech silence. The duration of gaps between spoken words is precisely replicated by sending a 'silence duration' value in the first packet of each new talk spurt.

30   This method has the advantage that it minimizes the delay when the network is not congested, adapts itself to operate optimally in various network conditions, without any user reconfiguration, and it does not require any

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complicated and specialized node handling of voice
packets (e.g. time-stamping).

     The invention also provides an apparatus for playing
out packets received through a packet network subject to
5 variable transmission delays, comprising a buffer for
receiving incoming packets; a speech reconstituter for
receiving said packets from said buffer and
reconstituting speech samples therefrom; a variable delay
unit for a applying a delay to the incoming packets; a
10 control unit for controlling said delay to apply a first
delay to an incoming sequence of packets and a variable
delay to subsequent packets of the sequence so as to
produce an appropriate constant play-out rate to said
buffer; means for monitoring the utilization of said
15 buffer; and means for adjusting said first delay applied
to the packets of a following sequence of packets to
maintain said fill level within desired limits to
minimize the risk of said buffer underflowing or
overflowing.

20   The invention will now be described in more detail,
by way of example only, with reference to the
accompanying drawings, in which:-

     Figure 1 is a schematic diagram showing a variable
transmission delay packet voice network ;

25   Figure 2 is a timing diagram of a packetized voice
transmission system in accordance with the invention; and

     Figure 3 is a block diagram of a variable delay
compensation apparatus in accordance with the invention.

     As shown in Figure 1, a speech input is transmitted
30 in packetized form through a packet network 2, for
example an ATM or Frame Relay network, which introduces a
variable delay during transmission through the network.

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The incoming packets are received by receiver 3, which outputs a re-assembled speech signal.

The packet network 2 introduces a variable propagation delay Δ. Receiver 3 introduces a further
5   delay δ in the manner to be described.

Referring now to Figure 2, input speech consists of spurts 4 separated by periods of silence 5. Each spurt 4 is represented by a sequence of packets 6, which when they are transmitted are separated by fixed spaces 7 as
10   shown at line 8. However, after transmission through the network the packets are no longer equally spaced, as shown at line 9, due to the variable propagation delays in the network. In accordance with the invention, as shown at line 10, the first packet 6a of each sequence is
15   subjected to a predetermined delay, which is estimated to be adequate to avoid buffer underflow and overflow. The remaining packets of the sequence are subjected to variable delays to maintain the appropriate constant output 11. This is then decoded to reproduce the initial
20   speech as shown at line 12.

Referring now to Figure 3, the incoming speech packets 6 are fed to variable delay unit 20, which introduces a variable delay between the packets. The output of variable delay unit 20 is fed to speech play-
25   out buffer 21, which outputs the speech packets to speech reconstituter 22, which turns the speech packets into constant rate speech samples, normally at 8KHz. These speech samples are then converted into analog speech signals in digital-to-analog converter 23.

30   Incoming speech packets 6 are also fed to packet analyzer 24 whose function is to identify the start of a speech spurt and trigger control unit 25, which sets the delay introduced by the variable delay unit 20 so as to produce a constant output rate. Buffer fill level

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monitor 26 monitors the fill level of speech play-out
buffer 21. Depending on the fill level of buffer 21,
control unit 25 varies the initial delay for the start of
the next talk spurt. Monitor 26 can be replaced by a
5   similar unit monitoring the dwell time of the packets in
the buffer. Alternatively, the buffer utilization can be
determined by monitoring the arrival rate of the packets.

In operation, the control unit 25 applies a pre-
determined delay to the first packet of a talk-spurt
10   detected by packet analyzer 24. The control unit 25 then
delays subsequent packets by an amount appropriate to
produce a constant play-out rate to the speech play-out
buffer 21. If the minimum number of packets in the buffer
is large (i.e., the buffer never becomes close to under-
15   flowing), the control unit 25 slowly reduces the build-
out delay. If packets arrive late, (i.e., the buffer
risks under-flowing), the build-out delay is increased in
order to minimize packet loss. Adjustment of the build-
out delay can be determined in a number of ways, such as
20   monitoring the minimum, maximum or average utilization of
the buffer. Alternatively, it is possible to monitor the
time a packet spends in the buffer.

Preferably, the delay adjustment is done during
speech silence. Normally, the duration of gaps between
25   spoken words is precisely replicated by sending a silence
duration value in the first packet of each new talk
spurt, which can be detected by the packet analyzer 24.
when the build-out delay has to be increased, the silence
duration is artificially increased. The result is a
30   larger build-out delay during the next talk spurt. When
the build-out delay has to be decreased, the silence
duration is artificially decreased. The result is a
shorter build-out delay during the next talk spurt.

Although the preferred embodiment has been described
35   with reference to audio signal, the invention may also be

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applied to video transmission. In this case video packets
carry the video data, and the speech reconstituter is
replaced by a video reconstituter, which operates in an
analogous manner. Indeed the invention is applicable to
5   any digitized physical signal that is transmitted in
packetized format and then reconstituted at the far end.
The video implementation looks the same as the
implementation shown in the drawings with the word
"video" substituted for the word "speech" throughout.

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Claims:

1.   A method of playing out packets received through a packet network subject to variable transmission delays, characterized in that it comprises the steps of:

      a) receiving incoming packets in a buffer;

      b) applying a delay to the first packet of a sequence of packets;

      c) applying a variable delay to subsequent packets of the sequence to produce an appropriate constant play-out rate of said packets to reproduce the desired output;

      d) monitoring the utilization of said buffer; and

      e) adjusting the delay applied to the first packet of a following sequence of packets to maintain said buffer utilization within desired limits to minimize the risk of said buffer underflowing or overflowing.

2.   A method as claimed in claim 1, characterized in that in step d the fill level of said buffer is monitored.

3.   A method as claimed in claim 1, characterized in that in step d the dwell time of said packets in said buffer is monitored.

4.   A method as claimed in claim 1, characterized in that said packets are voice or audio packets.

5.   A method as claimed in any of claims 1 to 4, characterized in that each said sequence represents a signal spurt.

6.   A method as claimed in claim 5, characterized in that said delay adjustment is carried out during signal activity in the gaps between signal spurts.

7.   A method as claimed in claim 5, characterized in that a signal inactivity duration value is inserted in the first packet of each spurt.

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8.   A method as claimed in claim 1, characterized in
that said packets are video packets.

9.   An apparatus for playing out packets received
through a network subject to variable transmission
delays, comprising:
        a) a buffer for receiving incoming packets;
        b) a signal reconstituter for receiving said packets
from said buffer and reconstituting signal samples
therefrom;
        c) a variable delay unit for applying a delay to the
incoming packets;
        d) a control unit for controlling said delay to
apply a first delay to an incoming sequence of packets
and a variable delay to subsequent packets of the
sequence so as to produce an appropriate constant play-
out rate to said buffer;
        e) means for monitoring the utilization of said
buffer; and
        f) means for adjusting said first delay applied to
the packets of a following sequence of packets to
maintain said buffer utilization within desired limits to
minimize the risk of said buffer underflowing or
overflowing.

10.   An apparatus as claimed in claim 9, characterized in
that said means for monitoring the utilization of said
buffer comprises a buffer fill level monitor.

11.   An apparatus as claimed in claim 9, characterized in
that said means for monitoring the utilization of said
buffer comprises a monitor determining the amount of time
the packets spend in the buffer.

12.   An apparatus as claimed in claim 9, characterized in
that it further comprises a packet analyzer for detecting
the start of a signal spurt.

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13. An apparatus as claimed in claim 7, characterized in that said delay adjustment is carried out during the gaps between signals to be transmitted.

14. An apparatus as claimed in claim 9, characterized in that said packets are voice or audio packets.

15. An apparatus as claimed in claim 9, characterized in that said packets are video packets.
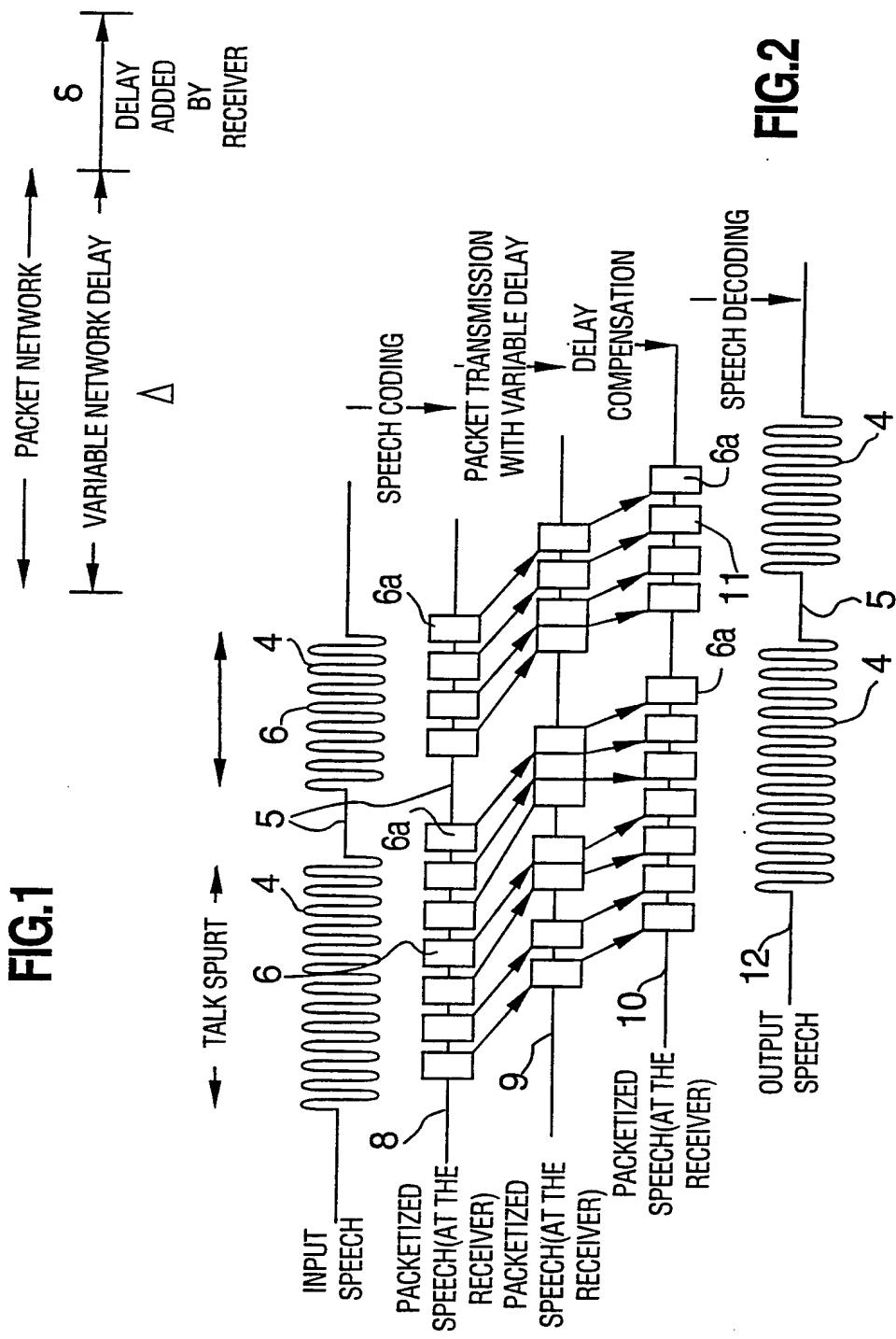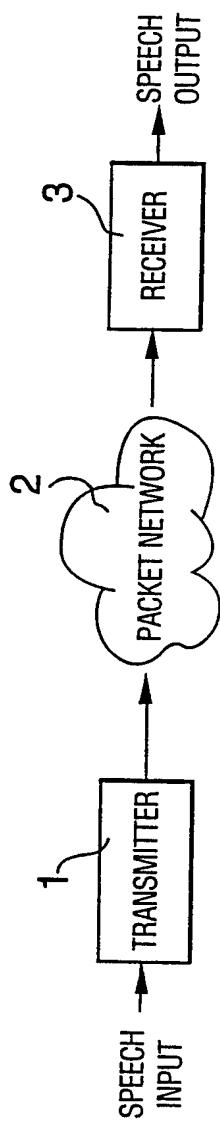
FIG.1

FIG.2
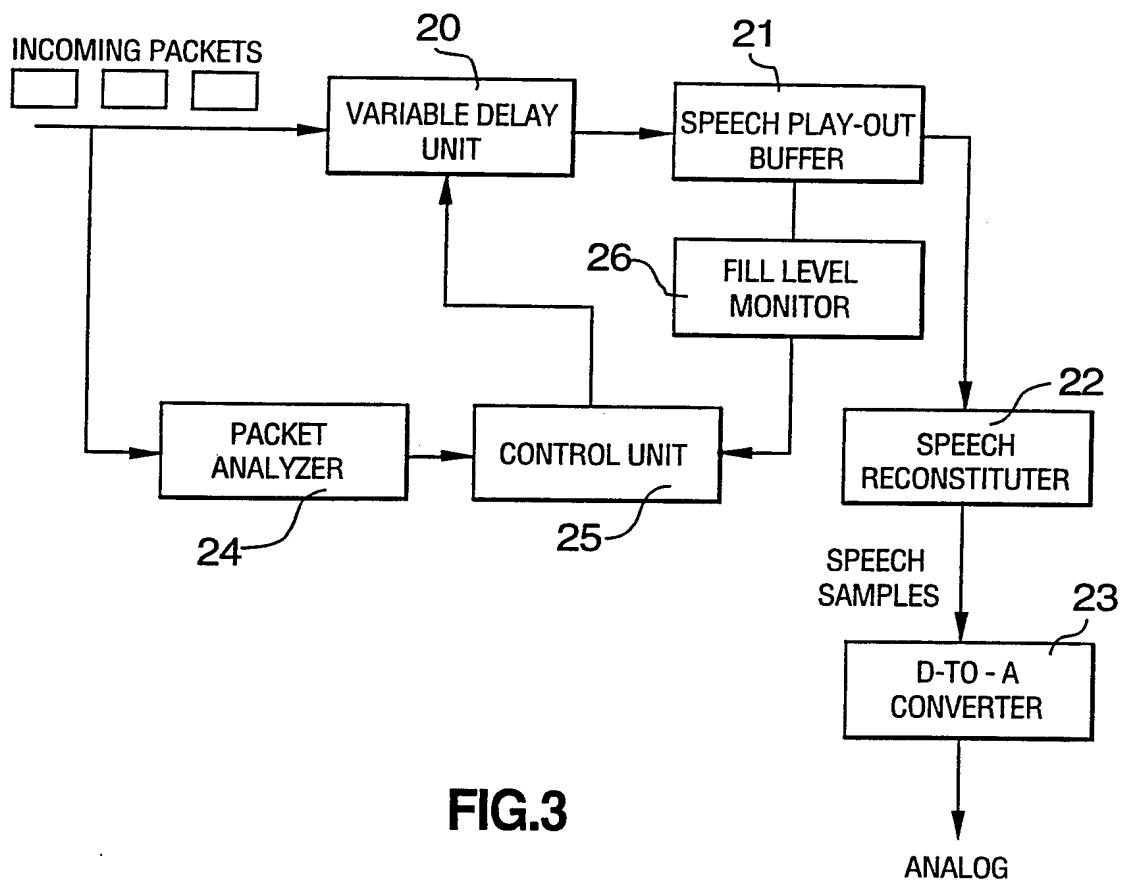
FIG.3

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6   H04Q11/04   H04J3/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6   H04Q   H04J   H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, DEC. 1983, USA, vol. SAC-1,no. 6, December 1983 ISSN 0733-8716, pages 1022-1028, MONTGOMERY W A 'Techniques for packet voice synchronization' see page 1022, right column, paragraph 3 - page 1023, left column, paragraph 4	1-6,8
A	see page 1023, right column, paragraph 7 - page 1024, left column, paragraph 1 see page 1024, right column, paragraph 3 see page 1026, left column, paragraph 6 - page 1027, right column, paragraph 1; figures 1,5,6	9-15

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X	Further documents are listed in the continuation of box C.

X	Patent family members are listed in annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
18 May 1995	8. 06. 95

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax (+31-70) 340-3016	Pieper, T

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INTERNATIONAL SEARCH REPORT

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category°	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP,A,0 130 431 (IBM) 9 January 1985 see page 11, line 16 - page 12, line 7 see page 14, line 31 - page 16, line 20; figures 2,3 ---	1,2,4,5, 9,10,14
A	WO,A,87 01254 (REPUBLIC TELCOM SYSTEMS) 26 February 1987 see page 7, line 6 - line 11 see page 19, line 31 - page 21, line 1; figure 4 ---	1,9
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